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IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

Re Application of: Ramo et al. : Attorney Docket No.: 944-003.191

Serial No.: 10/692,291 : Examiner: James S. Wozniak

Filed: October 23, 2003 : Art Unit: 2626

For: METHOD AND SYSTEM FOR PITCH CONTOUR QUANTIZATION IN AUDIO CODING

Mail Stop Appeal Brief - Patents
Commissioner for Patents
P.O. Box 1450
Alexandria, VA 22313-1450

RESPONSE TO NOTICE OF NON-COMPLIANT APPEAL BRIEF

Sir:

In response to the Notice of Non-Compliant Appeal Brief mailed August 11, 2006 for the present application, please accept the enclosed corrected Brief of the Appellants. The brief has now been revised to comply with required corrections.

The Examiner is invited to contact applicant's attorney if there are any questions.

Respectfully submitted,

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Cathy Sturmer 9.7.06
Date

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Commissioner for Patents
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Alexandria, VA 22313-1450

BRIEF OF APPELLANTS (37 CFR §41.37)

Sir:

This is an appeal from the final rejection contained in a Final Office Action mailed on November 22, 2005 (the "Final Office Action"), rejecting claims 1-24.

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9.7.06

Cathy Sturmer — Date

REAL PARTY IN INTEREST (37 CFR § 41.37(c)(1)(i))

The real party in interest in this action is Nokia Corporation, Keilalahdentie 4, FIN-02150 Espoo, Finland, by virtue of the Assignment dated November 10 and 14, 2003. The Assignment was recorded in the U.S. Patent and Trademark Office on February 9, 2004.

I. RELATED APPEALS AND INTERFERENCES (37 CFR § 41.37(c)(1)(ii))

There are no related appeals or interferences.

II. STATUS OF CLAIMS (37 CFR § 41.37(c)(1)(iii))

The status of the claims is:

Claims pending: 1-24.

Claims objected to: none.

Claims rejected: 1-24.

Claims on appeal: 1-24.

III. STATUS OF AMENDMENTS (37 CFR § 41.37(c)(1)(iv))

No amendment as to claims 3-15 and 17-24 has been filed subsequent to final rejection.

Claims 1, 2 and 16 have been amended in an amendment per 37 CFR §1.312, filed July 11, 2006.

V. SUMMARY OF CLAIMED SUBJECT MATTER (37 CFR § 41.37(c)(1)(v))

Appellants' invention is directed to a method and device for encoding an audio signal. Appellants' invention is also directed toward a computer software application having codes for carrying out the encoding method; a decoder for reconstructing an audio signal based on information provided by the encoding method; an electronic device having such a decoder; and a communication network having such a decoder.

The invention of claim 1 is directed to a method for encoding an audio signal for providing a plurality of parameters. The parameters include pitch contour data containing a plurality of pitch values representative of an audio segment.

In particular, the original pitch contour of the audio signal is divided into a plurality of pitch contour sub-segments. Each sub-segment has a start-point pitch value and an end-point pitch value. In the encoding process, each sub-segment is approximated by a linear contour segment defined by a start segment point and an end segment point. As shown in Figure 2, the pitch values of the original pitch contour is marked with x's and the linear contour segments are shown as straight lines. The start segment point and the end segment point of each linear segment are marked with circles. In order to obtain the linear contour segments, a plurality of simplified pitch contour segment candidates are generated. *See* Figure 4, step 504; Figure 5, step 604.

The deviations between the simplified pitch contour segment candidates and the pitch values in the corresponding sub-segments are then measured. *See* Figure 4, step 506; Figure 5, step 606.

Based on the measured deviations, the linear segments are selected for coding. *See* page 8, line 21 to page 9, line 17. In particular, the start segment point and the end segment point of a selected linear contour segment are generally different from the start-point pitch value and the end-point pitch value of the corresponding pitch contour sub-segment. *See* Figure 2.

In the invention of dependent claim 2, information indicative of the end points of the linear contour segments are provided to the decoder for audio signal reconstruction. *See* page 10, lines 17-29.

In the invention of dependent claim 3, the number of pitch values in some of the consecutive sub-segments is equal to or greater than 3. *See* the fourth, fifth and sixth segments in Figure 2.

In the invention of dependent claim 4, a pre-determined maximum value is used as a pre-selected condition in the generation of the simplified pitch contour segment candidates. *See* page 12, lines 18 – 22.

In the invention of dependent claim 5, a segment candidate is selected based on the maximum length among the segment candidates. *See* page 16, lines 13-15.

In the invention of dependent claim 6, a segment candidate is selected based on the minimum measured deviation among a group of candidates having the same length. *See* page 18, lines 12-15.

In the invention of dependent claim 7, the generation of segment candidates is carried out by adjusting the end segment point of the segment candidates. *See* page 9, lines 1-2.

In the invention of dependent claim 8, the audio signal comprises a speech signal. *See* page 10, lines 20–22.

In the invention of dependent claim 9, at least one of the selected candidates is a linear segment. *See* Figure 2.

In the invention of dependent claim 10, at least one of the selected candidates is a non-linear segment. *See* page 12, line 33 to page 13, line 6.

The invention of claim 11 is directed to an audio coding device having a data processing module for generating a plurality of pitch contour segment candidates based on the pitch contour data received via an input end. See Figure 3.

The processing module has an algorithm to measure the deviations between the generated pitch contour segment candidates and the pitch values in the corresponding sub-segments of the original pitch contour and to select a plurality of consecutive segment candidates based on the measured deviations and pre-selected criteria. *See* page 8, line 21 to page 9, line 17; page 10, lines 20-27.

In the invention of dependent claim 12, the coding device also has a quantization module for coding the pitch contour data corresponding to the selected segment candidates with characteristics of the selected segment candidate. *See* page 10, lines 27-29.

In the invention of dependent claim 13, the coding device also has a storage device for storing audio data indicative of the coded pitch contour data. *See* page 10, lines 27-29.

In the invention of dependent claim 14, the coding device has an output end connected to a storage medium for storing the coded pitch contour data. *See* Figure 3.

In the invention of dependent claim 15, the coding device has an output end for transmitting the coded pitch contour data to a decoder. *See* Figure 3.

The invention of claim 16 is directed to a computer software product having codes for carrying out the coding method as described in conjunction with claim 1 above. *See* page 10, lines 25-27.

The invention of claim 17 is directed to a decoder for reconstructing an audio signal based on information regarding the end points of the pitch contour sub-segments.

In particular, the decoder has an input end for receiving audio data. *See* Figure 3, block 40. The decoder also has a reconstruction module for reconstructing the audio segment based on the received audio data *See* page 10, lines 29-31.

In the invention of dependent claim 18, the audio data received by the decoder is recorded on an electronic media. *See* Figure 3.

In the invention of dependent claim 19, the audio data received by the decoder is transmitted through a communication channel. *See* Figure 3.

The invention of claim 20 is directed to an electronic device having a decoder as described in conjunction with claim 17. *See* Figure 3, block 50 and block 40.

In the invention of dependent claim 21, the audio data received by the decoder is recorded on an electronic media. *See* Figure 3.

In the invention of dependent claim 22, the audio data received by the decoder is transmitted through a communication channel. *See* Figure 3.

In the invention of dependent claim 23, the electronic device is a mobile terminal *See* Figure 3.

The invention of claim 24 is directed to a communication network having a plurality of base stations and a plurality of mobile stations, wherein at least one mobile station has a decoder as described in conjunction with claim 17.

The base stations and the mobile stations are depicted in Figure 6. The mobile station having a decoder is depicted in Figure 3. *See* block 50 and block 40.

VI. GROUNDS OF REJECTION TO BE REVIEWED ON APPEAL (37 CFR § 41.37(c)(1)(vi))

Claims 1-5, 7-12, 15, 17 and 20 are rejected under 35 U.S.C. 103(a) as being unpatentable over *Lee et al.* ("A very low bit rate speech coder based on a recognition/synthesis paradigm" IEEE Trans on Speech and Audio Processing, Vol. 9, No. 5, July 2001, hereafter referred to as *Lee*), in view of *Gao* (U.S. Patent No. 6,449,590).

In rejecting claim 1, the Examiner states that *Lee* discloses a method and system for improving coding efficiency having the following steps:

creating a plurality of simplified pitch contour segment candidates, each candidate corresponding to a sub-segment of the audio signal (Section V.A., pages 486-487);

measuring deviation between each of the simplified pitch contour segment candidates and the pitch values in the corresponding sub-segment; and

selecting a plurality of consecutive segment candidates to represent the audio segment (Section V.A., pages 486-487; Figure 5); and

coding the pitch contour data in the sub-segments of the audio signal corresponding to the selected segment candidates (Section V., page 486).

The Examiner admits that *Lee* fails to specifically suggest that the start and end points of a pitch contour sub-segment candidate may vary from the start and end points of the original speech sub-segment. The Examiner points to *Gao* for disclosing a means for time-warping the start and end points of a speech-sub-segment (col. 42, line 17 to col. 43, line 14).

The Examiner states that it would be obvious for one skilled in the art to modify the approximation method used by *Lee* using the time-warping method in *Gao* in order to implement an efficient pitch contour coding process capable of determining optimal start and ending times for a pitch contour segment (*Gao*, Col. 42, Line 9 – Col. 43, Line 14).

In rejecting claim 2, the Examiner states that *Lee* discloses providing information indicative of the end points so as to allow a decoder to reconstruct the audio signal in the audio segment based on the information instead of the pitch contour data (Section V.A., pages.486-487).

In rejecting claim 3, the Examiner states that *Lee* teaches pitch parameters per interval as shown in Figure 5.

In rejecting claim 4, the Examiner states that *Lee* teaches a maximum allowable contour approximation error threshold (Section V.A., pages 486-487).

In rejecting claim 5, the Examiner states that *Lee* teaches that if a longest candidate has an acceptable approximation error, no additional endpoints, which would result in shorter segments, are utilized (Section V.A., pages 486-487). The Examiner also states that *Lee* teaches coding based on the fewest number of endpoints needed to encode a pitch contour (section V-V.B., pages 486-488).

In rejecting claim 7, the Examiner points to *Gao* for disclosing the method for time warping and optimal endpoint determination of a pitch contour segment, as applied to Claim 1.

In rejecting claim 8, the Examiner states *Lee* teaches the linear estimation of a pitch contour as applied to claim 1, which is related to speech coding (Abstract).

In rejecting claim 9, the Examiner states that *Lee* shows a pitch contour approximation as a linear segment (Fig. 5).

In rejecting claim 10, the Examiner states that *Lee* teaches a B-spline approximation for a smoothed approximation of a pitch contour (Section V.A., page 487).

In rejecting claim 11, the Examiner states that *Lee* recites:

An input end for receiving the pitch contour data (Fig. 1); and

A data processing module, responsive to the pitch contour data, for creating a plurality of simplified pitch contour segment candidates, each candidate corresponding to a sub-segment of the audio signal, and wherein each sub-segment and candidate has a start-point pitch value and an end point pitch value and wherein the processing module comprises (F0 coding, Fig. 1; and Section V.A., pages 486-487):

An algorithm for measuring deviation between each of the simplified pitch contour segment candidates and said pitch values in the corresponding sub-segment; and an algorithm for selecting a plurality of consecutive segment candidates to represent the audio segment based on the measured deviations and one or more pre-selected criteria (error calculation and selecting candidates with error below a threshold amount, Section V.A., page 486-487; Fig. 5).

The Examiner admits that *Lee* fails to specifically suggest that start and end points of a pitch contour sub-segment candidate may vary from that of an original speech sub-segment, but points to *Gao* for disclosing a means for time warping start and end points of a speech-sub-segment (Col. 42, Line 17-Col. 43, Line 14).

The Examiner states that it would be obvious for one skilled in the art to modify the approximation method used by *Lee* using the time-warping method in *Gao* in order to implement an efficient pitch contour coding process capable of determining optimal start and ending times for a pitch contour segment (*Gao*, Col. 42, Line 9 – Col. 43, Line 14).

In rejecting claim 12, the Examiner states that *Lee* teaches a means for compressing an approximated pitch contour (Section V, page 486 and Section VI, page 488).

In rejecting claim 15, the Examiner states that *Lee* teaches the decoder having a concatenation unit as shown in Fig. 1.

In rejecting claim 17, the Examiner states that *Lee*, in view of *Gao*, teaches the pitch contour approximation encoder as applied to Claims 1 and 2, and that *Lee* also teaches the decoder as shown in Fig. 1 having a means for accepting data from an encoder and a unit concatenation module for reconstructing speech data.

In rejecting claim 20, the Examiner states that claim 20 contains subject matter similar to Claim 17, and thus, claim 20 is rejected for the same reasons.

Claim 6 is rejected under 35 U.S.C. 103(a) as being unpatentable over *Lee*, in view of *Gao*, and further in view of *Swaminathan et al.* (U.S. Patent Number 5,704,000, hereafter referred to as *Swaminathan*).

In rejecting claim 6, the Examiner states that *Lee*, in view of *Gao*, teaches the pitch contour approximation means as applied to Claim 4. The Examiner admits that *Lee*, in view of *Gao*, fails to specifically suggest comparing candidates having the same length and selecting the candidates with the minimum deviation, but points to *Swaminathan* for disclosing a means for selecting from a plurality of pitch candidates corresponding to pitch parameters of a specific pitch period (Col. 5, lines 14- 48).

The Examiner states that it would have been obvious to a person of ordinary skill in the art, at the time of invention, to modify the teachings of *Lee*, in view of *Gao*, with the means for comparing multiple pitch candidates to an original speech signal for a single pitch contour interval as disclosed by *Swaminathan* in order to further account for pitch estimation errors caused by spurious contaminants and distortion (*Swaminathan*, col. 5, Lines 23-31).

Claims 13, 14, 16, 18-19, and 21-24 are rejected under 35 U.S.C. 103(a) as being unpatentable over *Lee*, in view of *Gao*, and further in view of *Lumelsky* (U.S. Patent Number 6,246,672).

In rejecting claims 13 and 14, the Examiner states that *Lee*, in view of *Gao*, teaches the pitch contour approximation encoder as applied to Claims 11 and 12, and that *Lee* also teaches the transmission of encoded audio data to a decoder (Fig. 1; Section V, page 486,). The Examiner admits that *Lee*, in view of *Gao*, fails to disclose the storage of compressed audio data, but points to *Lumelsky* for disclosing a storage means coupled to an encoder for storing encoded audio data (Col. 6, Lines 32-56).

The Examiner states that it would have been obvious to a person of ordinary skill in the art, at the time of invention, to modify the teachings of *Lee*, in view of *Gao*, with the means for storing encoded audio data as taught by *Lumelsky* to implement a user-initialed means for playback of encoded audio information (*Lumelsky*, Col. 6, lines 39-43).

In rejecting claim 16, the Examiner states that *Lee*, in view of *Gao*, teaches the pitch contour approximation encoding method as applied to Claim 1, but fails to specifically suggest implementing that method as a computer program stored on a computer readable medium. The Examiner points to Lumelsky for disclosing storing an encoding method on a computer readable medium for the benefit of easily implementing an encoding method using a computer (*Lumelsky*, Col. 18, Lines 51-65).

In rejecting claims 18 and 19, the Examiner states that *Lumelsky* teaches the storage repository as applied to Claim 13 and further shows a wireless communication channel and decoder for decoding received audio data from the repository (Fig. 1).

In rejecting claims 21 and 22, the Examiner states that claims 21 and 22 contain subject matter similar to Claims 18 and 19, and thus, are rejected for the same reasons.

In rejecting claim 23, the Examiner states that *Lumelsky* further teaches a mobile user terminal connected to a wireless network (Fig. 1).

In rejecting claim 24, the Examiner states that *Lee*, in view of *Gao*, teaches the speech coding system utilizing piecewise linear approximation as applied to Claim 17. The Examiner admits that *Lee*, in view of *Gao*, fails to specifically disclose a communication network having a

plurality of base and mobile stations, but points to *Lumelsky* for disclosing such a communication network (base stations and mobile terminals, Col. 8, Lines 17-60).

VII. ARGUMENT (37 CFR § 41.37(c)(1)(vii))

A. THE CITED REFERENCES

1) The speech coding process according to *Lee*

Lee discloses an audio encoding method wherein successive linear approximation is used to substitute non-linear contour segments with linear contour. From the original fundamental frequency (FO) pitch contour, *Lee* picks two contour points as end points of a straight line. *See* Figure 6. Additional points on the pitch contour are then added to the line to form a new approximated contour such that the error between the approximated and original contours is maximum. *See* page 487, left column, second paragraph. This procedure is repeated until the error between the approximated contour and the original contour is smaller than a predetermined error value, d_{max}*. In the successive linear approximation according to *Lee*, all the points on the approximated contour are selected from the points on the original FO pitch contour. Thus, each of the linear segments of the approximated contour is bounded by two points of the original pitch contour.

Lee starts out using a frame as the basic synthesis unit because frame selection does not require a time-warping process. As such, synthesis of the speech signal can be carried out without time scale modification. *See* page 483, left column, fourth paragraph. Time scale modification may introduce artifacts. *See* page 483, left column, third paragraph. Because of the nature of successive linear approximation, *Lee's* FO coding is contour-wise rather than frame-wise. *See* page 486, left column, last paragraph.

2) The speech coding process according to *Gao*

Gao discloses a method of speech processing wherein the encoder chooses a start point and an end point from the original pitch lag contour for generating a linear contour segment. Time warping is then applied to the pitch lag contour segment between the start point and the end point so that the time-warped pitch lag contour segment matches the generate linear contour segment. In particular, the start point is previous pitch lag and the end point is a current pitch lag,

and time warping is carried out by a long-term pre-processor to temporally deform the weighted speech signal between the start and end points in order to conform to the generated pitched lag contour. *See* col.70, lines 47-53; col. 71, lines 10 – 17. *Gao* generates a linear pitch lag contour in a frame or sub-frame from previous and current pitch lag values. *See* Abstract. For example, on the original contour 811 during the time period between marker 813 and marker 817 (Figure 8a – 8c), *Gao* uses the pitch lag value at marker 813 and the pitch lag value at marker 815 as two end points for a linear pitch contour 831. Likewise, *Gao* uses the pitch lag value at marker 815 and the pitch lag value at marker 817 as two end points for another linear pitch contour 833. In order to match the contour defined by linear segments 831 and 833, the original pitch contour 811 is effectively compressed during some periods, e.g. at a time period 835, and expanded during others, e.g. at time period 837. *See* col. 42, lines 20-27.

Contrary to the assertion by the Examiner in rejecting claims 1, 11, 17, 20 and 24, *Gao* does not disclose or even suggest warping the start point and the end point of the generated speech contour. *Gao* only applies time warping on the weighted speech signal between the start and end points.

Furthermore, according to *Gao*, the encoder performs high-pass filtering and applies a perceptual weighting filter for providing the weighted speech signal, and a pitch preprocessing operation is applied to warp the weighted speech signal in order to match the interpolated pitch values that will be generated by the decoder. *See* col.5, lines 52 to 65; Figure 2. Thus, *Gao* uses high-pass filtering, perceptual weighting and speech signal warping to support lower bit-rate encoding modes. All these three steps are necessary to produce a linear pitch lag contour (Figure 8c) from a non-linear pitch lag contour (Figure 8b).

3) <u>The similarity between the approaches in *Lee* and *Gao*</u>

In *Lee's* coding method, the end points of a generated linear segment are points on the original FO contour. In *Gao*'s coding method, the end points of a generated linear segment are pitch lag values of the original pitch contour.

4) <u>The difference between the claimed invention and the approaches by *Lee* and *Gao*</u>

The end points of a generated linear segment according to *Lee* or *Gao* are points on the original contour. In contrast, at least some of end points of the linear contour segments are not points on the original contour.

5) <u>The differences between the approaches in *Lee* and *Gao*</u>

Lee's coding method avoids time warping because time scale modification may introduce artifacts. *Gao* relies totally on time warping. *Lee* generates an approximated contour from an original contour based on the variation of the original contour and thus the end points of a generated linear segment are generally unpredictable. Thus, *Lee's* coding method is contour-wise rather than frame-wise. *Gao* can choose the end points of any generated linear contour and *Gao's* coding method is frame-wise or subframe-wise (col. 42, lines 17 – 34). *Gao* uses high-pass filtering of the speech signal and a perceptual weighting filter for providing a weighted speech signal. *Lee* does not use those steps.

B. CLAIM REJECTIONS

<u>1) Claims 1, 11, 17 and 20</u>

i) <u>There would be no motivation or incentive to combine the two different approaches in *Lee* and *Gao*</u>

In order to raise a 103 rejection, the Examiner must show why a person skilled in the art would want to apply the method as disclosed in *Gao* to the method as disclosed in *Lee*. The Examiner fails to show such motivation. The Examiner simply states that modifying the teaching of *Lee* with means for time warping the start and end points of a speech sub-segment as taught by *Gao* can achieve an efficient pitch contour coding process capable of determining optimal start and ending times for a pitch contour segment (*Gao*, Col. 42, lines 9 – Col. 43, line 14).

There are many reasons why there would be no motivation to combine the approaches in *Lee* and *Gao*.

A. Complexity

As admitted by the Examiner, *Lee* uses the following four steps for pitch contour coding:

1) creating a plurality of simplified pitch contour segment candidates, each candidate corresponding to a sub-segment of the audio signal (Section V.A., pages 486-487);

2) measuring deviation between each of the simplified pitch contour segment candidates and the pitch values in the corresponding sub-segment; and

3) selecting a plurality of consecutive segment candidates to represent the audio segment (Section V.A., Pages 486-487; Figure 5); and

4) coding the pitch contour data in the sub-segments of the audio signal corresponding to the selected segment candidates (Section V. Page 486).

Gao uses a different approach in pitch contour coding. *Gao*'s process involves at least three steps (col.5, line 52 – 64):

a) high-pass filtering the speech signal;

b) applying a perceptual weighting filter to the high-pass filtered speech signal for providing a weighted speech signal, and

c) warping the weighted speech signal in order to match the interpolated pitch values that will be generated by the decoder.

None of the steps in *Gao* are used in *Lee*. Thus, in order to combine the method as disclosed in *Gao* to the method as disclosed in *Lee*, one must use all of the seven steps as shown above. The combined method requires a very complex encoder.

B. Compatibility

As mentioned earlier, *Lee's* coding method is different from *Gao*'s coding method in many aspects. It is uncertain whether it is possible to combine the time warping approach as used in *Gao* with the successive linear approximation as disclosed in *Lee*. For example, it is crucial for *Lee* to add a point in an approximated linear contour such that the error between the new approximated linear contour and the original contour is maximum so as to guarantee that the approximation error in the resulting approximated contour is below a pre-determined value. *See* p.487, left column, second paragraph. It is uncertain whether *Lee* can use the weighted speech signal to achieve the same goal.

C. *Lee* alone can accomplish what the combination of *Gao* and *Lee* may provide

Gao uses a time-warping method to replace non-linear pitch lag contour segments with linear pitch lag contour segments. The objective is to lower the coding bit-rate so as to meet a certain encoding mode.

Lee alone can lower the coding bit rate to meet a certain encoding mode by changing d_{max}*. For example, if a high bit-rate is available, *Lee* may use a smaller d_{max}* to improve the encoding accuracy. But when a lower bit-rate is required, *Lee* can use a larger d_{max}* in order to reduce the number of linear pitch contour segments. There is no need to introduce three additional steps as required in *Gao*.

D. Combining *Gao's* approach with *Lee* changes the principle of operation of *Lee*

The time duration between the end points in *Lee's* successive linear approximation changes with the variations of the original pitch contour because *Lee* selects points in the original contour such that the error between the approximated contour and the original contour is maximum. This maximum error is crucial for *Lee* to guarantee that the approximation error in the resulting contour is below d_{max}*.

The time duration between end points in *Gao*'s time warping method can be pre-selected. If the pre-selected time duration between end points as used in *Gao* is applied to *Lee*'s successive linear approximation, *Lee* may not be able to render the error between the approximated contour and the original contour being maximum. This is because pre-selecting time duration does not and cannot take into consideration the variations of the pitch contour at a time instant. Thus, by introducing the pre-selected time duration between end points, *Lee* may not be able to use its original principle of operation.

If the proposed modification or combination of the prior art would change the principle of operation of the prior art invention being modified, then the teachings of the reference are not sufficient to render the claims prima facie obvious. *See* MPEP, 2143.01(VI).

E. *Gao's* approach is not beneficial to the present invention

The present invention can lower the coding bit-rate to meet a certain encoding mode by changing the predetermined error value in the comparison step **508** as shown in the flowchart **500** of Figure 4 (page11, lines 17 - 24). The present invention does not use the time warping

technique as disclosed by *Gao*. There is no need to use the time warping technique as disclosed in *Gao* when a simple algorithm can be used to measure the deviation between a simplified pitch contour segment candidate and the pitch values in the original pitch contour for linear segment selection.

ii) <u>*Lee*, in view of *Gao*, fails to render the present invention obvious</u>

In sum, *Lee* does not require the approach as used in *Gao* in order to meet a certain bit-rate requirement. The present invention does not require the approach as used in *Gao* in order to meet a certain bit-rate requirement. *Lee* and *Gao* may not be compatible to each other. Even if they are compatible, the combination of *Lee* and *Gao* yields an unnecessary complex encoding system. The Examiner fails to show why a person skilled in the art would choose such a complex encoding system when a much simpler encoding system can achieve the same result.

Furthermore, *Lee* fails to disclose that the end points of the approximated linear segments are different from the original contour points, as admitted by the Examiner. *Gao* does not disclose or even suggest that the end points of the generated linear segment between any time duration are different from the start and end pitch lag values of the corresponding original pitch lag contour.

In contrast, at least some of the end points of the linear contour segments in the claimed invention are different from the start and end pitch values of the corresponding pitch contour.

For the above reasons, it is respectfully submitted that *Lee*, in view of *Gao*, does not render the invention as claimed in claims 1, 11, 17 and 20 obvious.

2) <u>Claims 2-5, 7-10, 12 and 25</u>

As for claims 2-5, 7-10, 12 and 15, they are dependent from claims 1 and 11 and recite features not recited in claims 1 and 11. For reasons regarding claims 1 and 11 above, it is respectfully submitted that claims 2-5, 7-10, 12 and 15 are also distinguishable over the cited *Lee* and *Gao* references.

3) <u>Claim 6</u>

Claim 6 is rejected under 35 U.S.C. 103(a) as being unpatentable over *Lee*, in view of *Gao* and further in view of *Swaminathan*. The Examiner cites *Swaminathan* for disclosing a

means for selecting from a plurality of pitch candidates corresponding to pitch parameters of a specific pitch period.

It is respectfully submitted that claim 6 is dependent from claim 1 and recites features not recited in claim 1. For reasons regarding claim 1 above, claim 6 is also distinguishable over the cited *Lee*, *Gao* and *Swaminathan* references.

4) <u>Claims 13, 14, 16, 18, 19 and 21-24</u>

Claims 13, 14, 16, 18, 19 and 21-24 are rejected under 35 U.S.C. 103(a) as being unpatentable over *Lee*, in view of *Gao* and further in view of *Lumelsky*. The Examiner cites *Lumelsky* for disclosing a storage means for storing encoded audio data.

It is respectfully submitted that claims 13, 14, 16, 18, 19 and 21-23 are dependent from claims 11, 17 and 20 and recite features not recited in claims 11, 17 and 20. For reasons regarding claims 11, 17 and 20 above, claims 13-14, 16, 18, 19 and 21-23 are also distinguishable over the cited *Lee*, *Gao* and *Lumelsky* references.

5) <u>Claim 24</u>

As for claim 24, it claims a communication network comprising a decoder as claimed in claim 17. For reasons regarding claim 17 above, it is respectfully submitted that claim 24 is also distinguishable over the cited *Lee*, *Gao* and *Lumelsky* references.

VIII CLAIMS APPENDIX (37 CFR § 41.37(c)(1)(viii))

1. A method for improving coding efficiency in audio coding, wherein an audio signal is encoded for providing parameters indicative of the audio signal, the parameters including pitch contour data containing a plurality of pitch values representative of an audio segment in time, said method comprising:

creating, based on the pitch contour data, a plurality of simplified pitch contour segment candidates, each candidate corresponding to a sub-segment of the audio signal, wherein each sub-segment has a start-point pitch value and an end-point pitch value and each candidate has a start segment point and an end segment point, and wherein the start segment points of at least some candidates are different from the start-point pitch values of the corresponding sub-segments and the end segment points of at least some candidates are different from the end-point pitch values of the corresponding sub-segments;

measuring deviation between each of the simplified pitch contour segment candidates and said pitch values in the corresponding sub-segment;

selecting, among said candidates, a plurality of consecutive segment candidates to represent the audio segment based on the measured deviations and one or more pre-selected criteria; and

coding the pitch contour data in the sub-segments of the audio signal corresponding to the selected segment candidates with characteristics of the selected segment candidates.

2. The method of claim 1, wherein the pitch contour data in the audio segment in time is approximated by a plurality of selected candidates, corresponding to a plurality of consecutive sub-segments in said audio segment, each of said plurality of selected candidates defined by a first end point and a second end point, and wherein said coding comprises providing information indicative of the end points so as to allow a decoder to reconstruct the audio signal in the audio segment based on the information instead of the pitch contour data.

3. The method of claim 1, wherein the number of pitch values in some of the consecutive sub-segment is equal to or greater than 3.

4. The method of claim 1, wherein said creating is limited by a pre-selected condition such that the deviation between each of the simplified pitch contour segment candidates and each of said pitch values in the corresponding sub-segment is smaller than or equal to a pre-determined maximum value.

5. The method of claim 4, wherein the created segment candidates have various lengths, and said selecting is based on the lengths of the segment candidates, and the pre-selected criteria include that

the selected candidate has the maximum length among the segment candidates.

6. The method of claim 4, wherein said selecting is based on the lengths of the segment candidates, and the pre-selected criteria include that

the measured deviation is minimum among a group of the candidates having the same length.

7. The method of claim 1, wherein said creating is carried out by adjusting the end segment point of the segment candidates.

8. The method of claim 1, wherein the audio signal comprises a speech signal.

9. The method of claim 2, wherein at least one of the selected candidates is a linear segment.

10. The method of claim 2, wherein at least one of the selected candidates is a non-linear segment.

11. A coding device for encoding an audio signal comprising pitch contour data containing a plurality of pitch values representative of an audio segment in time, said coding device comprising:

an input end for receiving the pitch contour data; and

a data processing module, responsive to the pitch contour data, for creating a plurality of simplified pitch contour segment candidates, each candidate corresponding to a sub-segment of the audio signal, wherein each sub-segment has a start-point pitch value and an end-point pitch value and each candidate has a start segment point and an end segment point, and wherein the start segment points of at least some candidates are different from the start-point pitch values of the corresponding sub-segments and the end segment points of at least some candidates are different from the end-point pitch values of the corresponding sub-segments, and wherein the processing module comprises:

an algorithm for measuring deviation between each of the simplified pitch contour segment candidates and said pitch values in the corresponding sub-segment; and

an algorithm for selecting, among said candidates, a plurality of consecutive segment candidates to represent the audio segment based on the measured deviations and pre-selected criteria.

12. The coding device of claim 11, further comprising

a quantization module, responsive to the selected segment candidates, for coding the pitch contour data in the sub-segments of the audio signal corresponding to the selected segment candidates with characteristics of the selected segment candidates.

13. The coding device of claim 12, wherein the quantization module provides audio data indicative of the coded pitch contour data in the sub-segments, said coding device further comprising

a storage device, operatively connected to the quantization module to receive the audio data, for storing the audio data in a storage medium.

14. The coding device of claim 12, further comprising an output end, operatively connected to a storage medium, for providing the coded pitch contour data to the storage medium for storage.

15. The coding device of claim 12, further comprising an output end for transmitting the coded pitch contour data to the decoder so as to allow the decoder to reconstruct the audio signal also based on the coded pitch contour data.

16. A computer software product embodied in an electronically readable medium for use in conjunction with an audio coding device, the audio coding device providing parameters indicative of the audio signal, the parameters including pitch contour data containing a plurality of pitch values representative of an audio segment in time, said software product comprising:

a code for creating a plurality of simplified pitch contour segment candidates based on the pitch contour data, each candidate corresponding to a sub-segment of the audio signal, wherein each sub-segment has a start-point pitch value and an end-point pitch value and each candidate has a start segment point and an end segment point, and wherein the start segment points of at least some candidates are different from the start-point pitch values of the corresponding sub-segments and the end segment points of at least some candidates are different from the end-point pitch values of the corresponding sub-segments, and;

a code for measuring deviation between each of the simplified pitch contour segment candidates and said pitch values in the corresponding sub-segment; and

a code for selecting, among said candidates, a plurality of consecutive segment candidates to represent the audio segment based on the measured deviations and pre-selected criteria, so as to allow a quantization module to code the pitch contour data in the sub-segments of the audio signal corresponding to the selected segment candidates with characteristics of the selected segment candidates.

17. A decoder for reconstructing an audio signal, wherein the audio signal is encoded for providing parameters indicative of the audio signal, the parameters including pitch contour data containing a plurality of pitch values representative of an audio segment in time, and wherein the pitch contour data in the audio segment in time is approximated by a plurality of consecutive simplified segments, each simplified segment corresponding to a sub-segment in the audio segment, wherein each of the sub-segments has a start-point pitch value and an end-point pitch value and each of the simplified segments is defined by a first end point and a second end point, and wherein the first end points of at least some simplified segments are different from the start-

point pitch values of the corresponding sub-segments and the second end points of at least some simplified segments are different from the end-point pitch values of the corresponding sub-segments, said decoder comprising:

an input for receiving audio data indicative of the end points defining the sub-segments; and

a reconstructing module, for reconstructing the audio segment based on the received audio data.

18. The decoder of claim 17, wherein the audio data is recorded on an electronic media, and wherein the input of the decoder is operatively connected to electronic media for receiving the audio data.

19. The decoder of claim 17, wherein the audio data is transmitted through a communication channel, and wherein the input of the decoder is operatively connected to the communication channel for receiving the audio data.

20. An electronic device comprising:

a decoder for reconstructing an audio signal, wherein the audio signal is encoded for providing parameters indicative of the audio signal, the parameters including pitch contour data containing a plurality of pitch values representative of an audio segment in time, and wherein the pitch contour data in the audio segment in time is approximated by a plurality of consecutive simplified segments, each simplified segment corresponding to a sub-segment in the audio segment, wherein each of the sub-segments has a start-point pitch value and an end-point pitch value and each of the simplified segments is defined by a first end point and a second end point, and wherein the first end points of at least some simplified segments are different from the start-point pitch values of the corresponding sub-segments and the second end points of at least some simplified segments are different from the end-point pitch values of the corresponding sub-segments, so as to allow the audio segment to be constructed based on the end points defining the simplified segments; and

an input for receiving audio data indicative of the end points and for providing the audio data to the decoder.

21. The electronic device of claim 20, wherein the audio data is recorded in an electronic medium, and wherein said input is operatively connected to the electronic medium for receiving the audio data.

22. The electronic device of claim 20, wherein the audio data is transmitted through a communication channel, and wherein the input is operatively connected to the communication channel for receiving the audio data.

23. The electronic device of claim 20, comprising a mobile terminal.

24. A communication network, comprising:

a plurality of base stations; and

a plurality of mobile stations communicating with the base stations, wherein at least one of the mobile stations comprises:

a decoder for reconstructing an audio signal, wherein the audio signal is encoded for providing parameters indicative of the audio signal, the parameters including pitch contour data containing a plurality of pitch values representative of an audio segment in time, and wherein the pitch contour data in the audio segment in time is approximated by a plurality of consecutive simplified segments, each simplified segment corresponding to a sub-segment in the audio segment, wherein each of the sub-segments has a start-point pitch value and an end-point pitch value and each of the simplified segments is defined by a first end point and a second end point, and wherein the first end points of at least some simplified segments are different from the start-point pitch values of the corresponding sub-segments and the second end points of at least some simplified segments are different from the end-point pitch values of the corresponding sub-segments; and

an input for receiving audio data indicative of the end points from at least one of the base stations for providing the audio data to the decoder.

IX. EVIDENCE APPENDIX (37 CFR § 41.37(c)(1)(ix))

There are no evidences submitted pursuant to 37 CFR §1.130, 1,131 or 1,132.

X. RELATED PROCEEDING APPENDIX (37 CFR § 41.37(c)(1)(x))

There are no prior decisions rendered by a court or the Board in any proceeding identified pursuant to 37 CFR § 41.37(c)(1)(ii).

CONCLUSION

It is respectfully submitted that the present invention as claimed is readily distinguishable over the cited *Lee, Gao, Swaminathan,* and *Lumelsky* references. Appellants' invention is not disclosed in the applied prior art and there is no fair basis for alleging that appellants' invention is obvious in regard to such art.

In view of the above, it is respectfully submitted that the rejection of claims 1-24 are in error and must be reversed. Such reversal is earnestly solicited.

Respectfully submitted,

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